Claims

1. A method for processing an audio signal, comprising operations of:

receiving a signal composed of a harmonic portion and a disturbance portion;

reducing an amplitude associated with the harmonic portion of the audio signal;

decreasing a sampling rate of the audio signal having the reduced amplitude of the harmonic portion;

identifying a type of signal sequence associated with the disturbance portion of the audio signal; and

modifying the disturbance portion according to the type of the signal sequence.

2. The method of claim 1, wherein the method operation of modifying the disturbance portion according to the type of the audio signal sequence includes,

removing the signal sequence when the type of the signal sequence is purely disturbance;

applying a frequency weighting factor to the signal sequence when the type of the signal sequence is purely harmonic; and

transforming the signal sequence to a frequency domain when the type of the signal sequence is a mixture of harmonic and disturbance signals.

3. The method of claim 2, wherein the method operation of removing the signal sequence when the type of the signal sequence is purely disturbance includes,

replacing the signal sequence through interpolation of both a signal preceding the signal sequence and a signal following the signal sequence.

4. The method of claim 2, wherein the method operation of applying a frequency weighting factor to the signal sequence when the type of the signal sequence is purely harmonic includes,

updating the frequency weighting factor for each frequency bin associated with the audio signal.

5. The method of claim 2, wherein the method operation of transforming the signal sequence to a frequency domain when the type of the signal sequence is a mixture of harmonic and disturbance signals includes,

scaling each frequency bin signal; and

transforming the scaled frequency bin signal to a time domain.

6. The method of claim 1, wherein the method operation of decreasing a sampling rate of the audio signal having the reduced amplitude of the harmonic portion includes,

downsampling the audio signal having the reduced amplitude by a factor of ten.

7. A method for reducing a noise disturbance associated with an audio signal received through a microphone, comprising operations of:

magnifying a noise disturbance of the audio signal relative to a remaining component of the audio signal;

decreasing a sampling rate of the audio signal;

applying an even order derivative to the audio signal having the decreased sampling rate to define a detection signal; and

adjusting the noise disturbance of the audio signal according to a statistical average of the detection signal.

8. The method of claim 7, wherein the method operation of magnifying a noise disturbance of the audio signal relative to a remaining component of the audio signal includes,

processing the audio signal through an inverse impulse response filter.

9. The method of claim 7, wherein the method operation of decreasing a sampling rate of the audio signal includes,

downsampling the audio signal by a factor of ten.

10. The method of claim 7, wherein the method operation of applying an even order derivative to the audio signal having the decreased sampling rate to define a detection signal further differentiates the noise disturbance of the audio signal from the remaining component of the audio signal.

11. The method of claim 7, wherein the method operation of adjusting the noise disturbance of the audio signal according to a statistical average of the detection signal includes,

identifying if a signal sequence associated with the noise disturbance includes the remaining component of the audio signal.

12. The method of claim 11, wherein if the signal sequence associated with the noise disturbance includes the remaining component of the audio signal, then the method includes,

transforming the audio signal to the frequency domain from a time domain;

scaling each frequency bin of the transformed audio signal according to a weighting factor to define a scaled audio signal; and

transforming the scaled audio signal back to the time domain.

13. The method of claim 11, wherein if the signal sequence associated with the noise disturbance is solely noise disturbance signal, then the method includes,

replacing the signal sequence through interpolation of both a signal preceding the signal sequence and a signal following the signal sequence.

14. A computer readable medium having program instructions for processing an audio signal, comprising:

program instructions for receiving a signal composed of a harmonic portion and a disturbance portion;

program instructions for reducing an amplitude associated with the harmonic portion of the audio signal;

program instructions for decreasing a sampling rate of the audio signal having the reduced amplitude of the harmonic portion;

program instructions for identifying a type of signal sequence associated with the disturbance portion of the audio signal; and

program instructions for modifying the disturbance portion according to the type of the signal sequence.

15. The computer readable medium of claim 14, wherein the program instructions for modifying the disturbance portion according to the type of the audio signal sequence includes,

program instructions for removing the signal sequence when the type of the signal sequence is purely disturbance;

program instructions for applying a frequency weighting factor to the signal sequence when the type of the signal sequence is purely harmonic; and

program instructions for transforming the signal sequence to a frequency domain when the type of the signal sequence is a mixture of harmonic and disturbance signals.

16. The computer readable medium of claim 15, wherein the program instructions for removing the signal sequence when the type of the signal sequence is purely disturbance includes,

program instructions for replacing the signal sequence through interpolation of both a signal preceding the signal sequence and a signal following the signal sequence.

17. The computer readable medium of claim 15, wherein the program instructions for applying a frequency weighting factor to the signal sequence when the type of the signal sequence is purely harmonic includes,

program instructions for updating the frequency weighting factor for each frequency bin associated with the audio signal.

18. The computer readable medium of claim 15, wherein the program instructions for transforming the signal sequence to a frequency domain when the type of the signal sequence is a mixture of harmonic and disturbance signals includes,

program instructions for scaling each frequency bin signal; and

program instructions for transforming the scaled frequency bin signal to a time domain.

19. The computer readable medium of claim 14, wherein the program instructions for decreasing a sampling rate of the audio signal having the reduced amplitude of the harmonic portion includes,

program instructions for downsampling the audio signal having the reduced amplitude by a factor of ten.

20. A computer readable medium having program instructions for reducing a noise disturbance associated with an audio signal received through a microphone, comprising operations of:

program instructions for magnifying a noise disturbance of the audio signal relative to a remaining component of the audio signal;

program instructions for decreasing a sampling rate of the audio signal;

program instructions for applying an even order derivative to the audio signal having the decreased sampling rate to define a detection signal; and

program instructions for adjusting the noise disturbance of the audio signal according to a statistical average of the detection signal.

21. The computer readable medium of claim 20, wherein the program instructions for magnifying a noise disturbance of the audio signal relative to a remaining component of the audio signal includes,

program instructions for processing the audio signal through an inverse impulse response filter.

22. The computer readable medium of claim 20, wherein the program instructions for decreasing a sampling rate of the audio signal includes,

program instructions for downsampling the audio signal by a factor of ten.

23. The computer readable medium of claim 20, wherein the program instructions for applying an even order derivative to the audio signal having the decreased sampling rate to define a detection signal further differentiates the noise disturbance of the audio signal from the remaining component of the audio signal.

24. The computer readable medium of claim 20, wherein the program instructions for adjusting the noise disturbance of the audio signal according to a statistical average of the detection signal includes,

program instructions for identifying if a signal sequence associated with the noise disturbance includes the remaining component of the audio signal.

25. The computer readable medium of claim 24, wherein if the signal sequence associated with the noise disturbance includes the remaining component of the audio signal, then the computer readable medium includes,

program instructions for transforming the audio signal to the frequency domain from a time domain;

program instructions for scaling each frequency bin of the transformed audio signal according to a weighting factor to define a scaled audio signal; and

program instructions for transforming the scaled audio signal back to the time domain.

26. The computer readable medium of claim 24, wherein if the signal sequence associated with the noise disturbance is solely noise disturbance signal, then the computer readable medium includes,

program instructions for replacing the signal sequence through interpolation of both a signal preceding the signal sequence and a signal following the signal sequence.

27. A system capable of canceling disturbances associated with an audio signal, comprising:

a computing device including logic for processing an audio signal, the logic for processing the audio signal including,

logic for generating a detection signal from the audio signal; and

logic for determining whether a signal sequence of the audio signal is a disturbance through analysis of a corresponding signal sequence of the detection signal;

an input device operatively connected to the computing device; and

a microphone configured to capture the audio signal, wherein the microphone is positioned so that a source of the disturbance is located within a near-field associated

with the microphone and a source of a target component of the audio signal is located within a far field associated with the microphone.

28. The system of claim 27, wherein the microphone is affixed to the input device.

29. The system of claim 27, wherein the logic for determining whether a signal sequence of the audio signal is a disturbance through analysis of a corresponding signal sequence of the detection signal includes,

logic for transforming the audio signal from a time domain to a frequency domain;

logic for adjusting a frequency bin of the audio signal in the frequency domain; and

logic for transforming the adjusted audio signal to the time domain from the frequency domain.

30. The system of claim 27, wherein the disturbance is a mechanical disturbance having a frequency range between about 0 and about 800 Hertz.

31. The system of claim 27, wherein the input device is a video game controller.

32. The system of claim 27, wherein the computing device is a video game console.

33. The system of claim 27, wherein each logic element is one of or a combination of software and hardware.

34. A video game controller, comprising:

a microphone affixed to the video game controller, the microphone configured to detect an audio signal that includes a target audio signal in a far field relative to the microphone and disturbance noise in a near field relative to the microphone;

logic configured to process the audio signal, the logic including,

detection signal logic configured to generate a detection signal through application of an even ordered derivative to the audio signal; and

disturbance cancellation logic configured to remove disturbance noise from the audio signal through analysis of the detection signal.

35. The video game controller of claim 34, wherein the disturbance cancellation logic includes,

logic for identifying if a signal sequence of the disturbance noise is associated with the target audio signal.

36. The video game controller of claim 35, further comprising multiple microphones, wherein each of the multiple microphones is configured to independently identify whether the disturbance noise is above a threshold level.

37. The video game controller of claim 34, wherein the detection signal logic includes,

downsampling logic configured to reduce an amount of data associated with the detection signal, as compared to the audio signal, by a factor of ten.

38. An integrated circuit, comprising:

circuitry configured to receive an audio signal from at least one microphone in a multiple noise source environment;

circuitry configured to perform signal decorrelation on the audio signal;

circuitry configured to downsample the decorrelated audio signal;

circuitry configured to apply a differentiation operation to the downsampled audio signal;

circuitry configured to detect a noise disturbance signal sequence within the differentiated audio signal; and

circuitry configured to remove a signal sequence of the audio signal associated with the noise disturbance signal sequence.

39. The integrated circuit of claim 38, wherein the circuitry configured to perform signal decorrelation on the audio signal is a linear prediction error filter.

40. The integrated circuit of claim 38, wherein the circuitry configured to downsample the decorrelated audio signal reduces an amount of data associated with the audio signal by a factor of ten.

41. The integrated circuit of claim 38, wherein the differentiation is a fourth order differentiation operation.

42. The integrated circuit of claim 38, wherein the circuitry configured to detect a noise disturbance signal sequence within the differentiated audio signal includes,

circuitry configured to identify whether the noise disturbance signal sequence includes a target signal sequence.

43. The integrated circuit of claim 38, wherein the circuitry configured to remove a signal sequence of the audio signal associated with the noise disturbance signal sequence includes,

circuitry configured to perform a linear interpolation based upon a previous signal sequence and a later signal sequence.

44. The integrated circuit of claim 38, wherein the integrated circuit is contained within one of a video game controller and a video game console.